Method for Encoding a Digital Signal into a Scalable Bitstream; Method for Decoding a Scalable Bitstream

Background of the Invention

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Recently, with the advances in computers, networking and
communications streaming audio contents over networks such as
the Internet, wireless local area networks, home networks and
commercial cellular phone systems is becoming a mainstream

10   means of audio service delivery. It is believed that with the
progress of the broadband network infrastructures, including
xDSL, fiber optics, and broadband wireless access, bit rates
for these channels are quickly approaching those for delivering
high sampling-rate, high amplitude resolution (e.g. 96 kHz, 24

15   bit/sample) lossless audio signals. On the other hand, there
are still application areas where high-compression digital
audio formats, such as MPEG-4 AAC (described in [1]) are
required. As a result, interoperable solutions that bridge the
current channels and the rapidly emerging broadband channels

20   are highly demanded. In addition, even when broadband channels
are widely available and the bandwidth constraint is ultimately
removed, a bit-rate-scalable coding system that is capable to
produce a hierarchical bit-stream whose bit-rates can be
dynamically changed during transmission is still highly

25   favorable. For example, for applications where packet loss
occurs occasionally due to accidents or resource sharing
requirements, the current broadband waveform representations
such as PCM (Pulse Code Modulation) and lossless coding formats
may suffer serious distortions in a streaming situation.

30   However, this problem can be solved if one could set packet
priorities in the case that network resources are dynamically
changing. Finally, a bit-rate-scalable coding system also

provides the server advantageous for audio streaming services,
where graceful QoS degradation could be achieved if an
excessive number of demands from client sites come.

5   Previously many lossless audio coding algorithms have been
proposed (see [2]-[8]). Most approaches rely on a prediction
filter to remove the redundancy of the original audio signals
while the residuals are entropy coded (as described in [5]-
[12]). Due to the existence of the predictive filters, the bit-
10   streams generated by these prediction based approaches are
difficult and not efficient (see [5],[6]), if not impossible,
to be scaled to achieve bit-rate scalability. Other approaches,
such as described in [3], build the lossless audio coder
through a two layer approach where the original audio signals
15   are first coded with a lossy encoder and its residual is then
lossless coded with a residual encoder. Although this two layer
design provides some sort of bit-rate scalability, its
granularity is too coarse to be appreciated by audio streaming
applications. Audio codecs that provide the fine grain
20   scalability on bit-rate were previously proposed in [4] and
[18], however, unlike the system to be discussed here, those
codecs don't provide the backward compatibility that the lossy
bit-streams produced by both codecs are incompatible to any
existing audio codec.
25

In [21], [22], [23] perceptual models are described.

The object of the invention is to provide a method for encoding
a digital signal in a scalable bitstream wherein backward
30   compatibility can be maintained.

Summary of the Invention

A method for encoding a digital signal into a scalable
bitstream is provided, which method comprises: quantizing the
digital signal, and encoding the quantized signal to form a

5   core-layer bitstream; performing an error mapping based on the
digital signal and the core-layer bitstream to remove
information that has been encoded into the core-layer bitstream,
resulting in an error signal; bit-plane coding the error signal
based on perceptual information of the digital signal,

10   resulting in an enhancement-layer bitstream, wherein the
perceptual information of the digital signal is determined
using a perceptual model; and
-multiplexing the core-layer bitstream and the enhancement-
layer bitstream, thereby generating the scalable bitstream.

15

Further, an encoder for encoding a digital signal into a
scalable bitstream, a computer readable medium, a computer
program element, a method for decoding a scalable bitstream
into a digital signal, a decoder for decoding a scalable

20   bitstream into a digital signal, a further computer readable
medium and a further computer program element according to the
method described above are provided.

In one embodiment, a lossless audio codec that achieves fine

25   grain bit-rate scalability (FGBS) with the following
characteristics is presented:
- Backward compatibility: a high-compression core-layer bit-
stream, such as MPEG-4 AAC bitstream, is embedded in the
lossless bit-stream.

30   - Perceptually embedded lossless bit-stream: the lossless bit-
stream can be truncated to any lossy rates without loss in the
perceptual optimality in the reconstructed audio.

- Low complexity: it adds only very limited calculation upon AAC (binary arithmetic codec) as well as very limited memory.

5   The abundant functionality provided by the presented audio codec suggests its capability of serving as a "universal" audio format to meet the various rate/quality requirements for different audio streaming or storage applications. For example, a compliant MPEG-4 AAC bit-stream which is used as the core-layer bitstream can be easily extracted from the bit-stream

10   generated using the codec for conventional MPEG-4 AAC audio services. On the other hand, lossless compression is also provided by the codec for audio editing or storage applications with lossless reconstruction requirement. In audio streaming applications, where the FGBS is needed, the lossless bit-stream

15   of the codec can be further truncated to lower bit-rates at the encoder/decoder or in the communication channel for any rate/fidelity/complexity constraints that may be arisen in practical systems.

20   In one embodiment a method for encoding a digital signal to form a scalable bitstream is provided, wherein the scalable bitstream can be truncated at any point to produce a lower quality (lossy) signal when decoded by a decoder. The method can be used for encoding any types of digital signal, such as

25   audio, image or video signals. The digital signal, which corresponds to a physical measured signal, may be generated by scanning at least a characteristic feature of a corresponding analog signal (for example, the luminance and chrominance values of a video signal, the amplitude of an analog sound

30   signal, or the analog sensing signal from a sensor). For example, a microphone may be used to capture an analog audio signal, which is then converted to a digital audio signal by

sampling and quantizing the captured analog audio signal. A
video camera may be used to capture an analog video signal,
which is then converted to a digital video signal using a
suitable analog-to-digital converter. Alternatively, a digital

5   camera may be used to directly capture image or video signal
onto an image sensor (CMOS or CCD) as digital signals.

The digital signal is quantized and coded to form a core-layer
bitstream. The core-layer bitstream forms the minimum bit-

10   rate/quality of the scalable bitstream.

An enhancement-layer bitstream is used to provide an additional
bit-rate/quality of the scalable bitstream. The enhancement-
layer bitstream is formed according to the invention by

15   performing an error mapping based on the transformed signal and
the core-layer bitstream to generate an error signal. The
purpose of performing error mapping is to remove the
information which has already been coded into the core-layer
bitstream.

20
The error signal is bit-plane coded to form the enhancement-
layer bitstream. The bit-plane coding of the error signal is
performed based on perceptual information, i.e. the perceived
or perceptual importance, of the digital signal. Perceptual

25   information used in this present invention refers to
information which is related to the human sensory system, for
example the human visual system (i.e. the human eye) and the
human auditory system (i.e. the human ear). Such perceptual
information for the digital signal (video or audio) is obtained

30   using a perceptual model, for example the Psychoacoustic Model
I or II in the MPEG-1 audio (described in [21]), for audio
signals, and the Human Visual System Model for image (described

in [22]), and the Spatio-Temporal Model used in video
(described in [23]).

The psychoacoustic model is based on the effect that the human
ear is only able to pick up sounds within a certain band of
frequencies depending on various environmental conditions.
Similarly, the HVM (human visual model) is based on the effect
that the human eye is more attentive to certain motion, colors
and contrast.

The core-layer bitstream and the enhancement-layer bitstream
are multiplexed to form the scalable bitstream.

The scalable bitstream can be decoded to losslessly reconstruct
the digital signal. As mentioned above, the core-layer
bitstream is an embedded bitstream which forms the minimum bit-
rate/quality of the scalable bitstream, and the enhancement-
layer bitstream forms the lossy to lossless portion of the
scalable bitstream. As the enhancement-layer bitstream is
perceptually bit-plane coded, the enhancement-layer bitstream
can be truncated, in a manner such that data in the
enhancement-layer bitstream which are less perceptually
important are truncated first, to provide perceptual
scalability of the scalable bitstream. In other words, the
scalable bitstream can be scaled by truncating the enhancement-
layer bitstream, so that the enhancement-layer bitstream, and
hence the scalable bit-stream, can be perceptually optimized
even when truncated to a lower bit-rate/quality.

The method according to the invention can be used as a lossless
encoder for digital signal, such as image, video or audio
signal, in high bandwidth or hi-fidelity systems. When the

bandwidth requirement changes, the bit-rate of the bitstream
generated by the encoder may be changed accordingly to meet the
change in bandwidth requirements. Such a method can be
implemented in many applications and systems such as MEG audio,

5   image and video compression of JPEG 2000.

According to an embodiment of the invention, the digital signal
is transformed into a suitable domain before being quantized to
form the quantized signal. The digital signal may be

10   transformed within the same domain, or from one domain to
another domain in order to better represent the digital signal,
and thereby to allow an easy and efficient quantizing and
coding of the digital signal to form the core-layer bitstream.
Such domain may include, but not limited to, the time domain,

15   the frequency domain, and a hybrid of the time and frequency
domains. The transformation of the digital signal may even be
carried out by a unitary matrix, I.

In one embodiment, the digital signal is transformed to a

20   transformed signal using an integer Modified Discrete Cosine
Transform (intMDCT). The intMDCT is a reversible approximation
to the Modified Discrete Cosine Transform (MDCT) filterbank,
which is commonly used in a MPEG-4 AAC coder. Other transforms
for transforming the digital signal into a suitable domain for

25   further processing can also be used, including, but not limited
to, Discrete Cosine Transform, Discrete Sine Transform, Fast
Fourier Transform and Discrete Wavelet Transform.

When intMDCT is used to transform the digital signal to the

30   transformed signal, the transformed signal (specifically the
intMDCT coefficients which describes the transformed signal) is
preferably normalized or scaled to approximate the output of a

MDCT filterbank. The normalizing of the intMDCT-transformed
signal may be useful in the case when a quantizer for
quantizing the transformed signal, for example an AAC quantizer,
has MDCT filterbank with a global gain different from the
5   global gain of the intMDCT filterbank. Such normalizing process
approximates the intMDCT-transformed signal to the MDCT
filterbank so that it is suitable to be directly quantized and
coded by the quantizer to form the core-layer bitstream.

10   For encoding an audio digital signal, the digital/transformed
signal is preferably quantized and coded according to the MPEG
AAC specification to generate the core-layer bitstream. This is
because AAC is one of the most efficient perceptual audio
coding algorithm for generating a low bit-rate but high quality
15   audio bitstream. Therefore, the core-layer bitstream generated
using AAC (referred as AAC bitstream) has a low bit-rate, and
even when the scalable bitstream is truncated to the core-layer
bitstream, the perceptual quality of the truncated bitstream is
still high. It should be noted that other quantization and
20   coding algorithms/methods, for example MPEG-1 Audio Layer 3,
(MP3) or other proprietary coding/quantizing methods for
generating the core-layer bitstream can also be used.

The error mapping which removes information which has already
25   been coded into the core-layer bitstream and which generates a
residual signal (or error signal) is performed by subtracting
the lower quantization threshold (closer to zero) of each
quantized value of the quantized signal from the transformed
signal. Such error mapping procedure based on quantization
30   threshold has the advantage that the values of the residual
signal is always positive, and the amplitude of the residual
signal is independent of the quantization threshold. This

allows a low-complexity and efficient embedded coding scheme to be implemented. It is however also possible to subtract a reconstructed transformed signal from the transformed signal to generate the residual signal.

5

To determine the perceptual information of the digital signal for bit-plane coding of the error signal, the psychoacoustic model can be used as the perceptual model. The psychoacoustic model may be based on Psychoacoustic Model I or II used in MPEG-1 audio (as described in [21]), or the Psychoacoustic Model in MPEG-4 audio (as described in [19]). When a perceptual quantizer, such as the one used according to AAC, is used for quantizing and coding the digital/transformed signal, the perceptual model used in the perceptual quantizer may also be used to determine the perceptual information for bit-plane coding of the error signal. In other words, a separate perceptual model is not needed in this case to provide the perceptual information for bit-plane coding of the error signal.

The perceptual information for bit-plane coding of the error signal is preferably also multiplexed with the core-layer and enhancement-layer bitstreams to form the scalable bitstream as side information. The side information can be used to reconstruct the error signal by a decoder.

The error signal is arranged in a plurality of bit-planes, with each bit-plane having a plurality of bit-plane symbols.

In an embodiment of the invention, the arrangement or order of the bit-planes of the error signal is changed or shifted, and the bit-planes are subsequently scanned and coded in a consecutive sequential manner. The bit-planes are shifted in a

way such that when the bit-plane coding is performed on the
shifted bit-planes, bit-planes comprising the more perceptually
important bit-plane symbols are scanned and coded first. In
this embodiment, all the bit-plane symbols in a bit-plane are
5 coded before coding the bit-plane symbols of a subsequent
adjacent bit-plane.

In another embodiment of the invention, the bit-plane symbols
of the bit-planes are scanned and coded in a sequence based on
10 the perceptual information. In other words, not all the bit-
plane symbols in a bit-plane are coded before coding the bit-
plane symbols from another bit-plane. The scanning and coding
sequence of the bit-plane symbols from the plurality of bit-
planes is determined based on the perceptual information such
15 that bit-plane symbols which are more perceptually important
are coded first.

The perceptual information of the digital signal determined by
the perceptual model may include the first (or maximum) bit-
20 plane $M(s)$ (i.e. a number (index) specifying the first bit-
plane) of the plurality of bit-planes for the bit-plane coding
of the error signal, and/or the Just Noticeable Distortion (JND)
level of the digital signal. It should be noted that the
perceptual information relates to the digital signal for every
25 different domain characteristics (for example frequency, time,
signal amplitude, etc) or a range of domain characteristics.
For example, when the digital signal is transformed to the
frequency domain, the perceptual information of the digital
signal at every frequency or in a band of frequency (frequency
30 band s, or more generally, domain band s) values may be
different, indicating that the signal may be more important
perceptually at certain frequencies.

In an embodiment of the invention, perceptual significance P(s)
of the digital signal, corresponding to each frequency band s,
is determined as the perceptual information. In this embodiment,
5   the JND level $\tau(s)$ of the digital signal corresponding to the
bit-plane of the error signal is determined. The bit-plane
corresponding to the JND level $\tau(s)$ is then subtracted from the
index of the first bit-plane of the plurality of bit-planes for
the bit-plane coding of the error signal M(s) to result in the
10   perceptual significance P(s). The perceptual significance P(s)
can be used for controlling the shifting of the bit-planes, so
that bit-planes comprising the more perceptually important bit-
plane symbols are scanned and coded first. More advantageously,
the perceptual significance P(s) can be used to control the
15   scanning and coding sequence of the bit-plane symbols from the
plurality of bit-planes so that bit-plane symbols which are
more perceptually important are coded first.

In a further embodiment of the invention, the perceptual
20   significance P(s) is normalized to form a normalized perceptual
significance Ps'(s). In this embodiment, a common perceptual
significance Ps_common of the digital signal is defined based
on a function of the perceptual significance Ps(s). Examples of
such function of the perceptual significance Ps(s) include the
25   average value, the maximum value, the minimum value or a
normalized value of the perceptual significance Ps(s). The
common perceptual significance Ps_common is subtracted from the
perceptual significance Ps(s) to result in the normalized
perceptual significance Ps'(s) for each frequency band s. When
30   the frequency band s contains at least one non-zero value
quantized signal, the frequency band s is a significant band.

Otherwise, the frequency band s is an insignificant band s. For significant band, the value of the corresponding perceptual significance Ps(s) is set to the value of the common perceptual significance Ps_common. For insignificant band, the

5   corresponding normalized perceptual significance Ps'(s) is multiplexed with the core-layer bitstream and the enhancement-layer bitstream for generating the scalable bitstream for transmission. This normalized perceptual significance Ps'(s) is transmitted in the scalable bitstream as side information for

10 decoding the scalable bitstream in a decoder.

The normalizing of the perceptual significance Ps(s) by defining a common perceptual significance Ps_common has the advantage of reducing the amount of perceptual information to

15 be transmitted in the scalable bitstream by utilizing information obtained when quantizing the digital/transformed signal to generate the core-layer bitstream. Therefore, perceptual information, in particular the normalized perceptual significance Ps'(s), is only needed to be transmitted to the

20 decoder side for insignificant band, as such perceptual information for significant band can be easily regenerated by the decoder.

The index of the first (or maximum) bit-plane of the plurality

25 of bit-planes for the bit-plane coding of the error signal M(s), which is part of the perceptual information of the digital signal, can be determined from the maximum quantization interval used for quantizing the digital/transformed signal. For significant band, the maximum quantization interval (the

30 difference between the higher and lower quantization threshold corresponding to each quantized value of the quantized signal) is determined, and the said first bit-plane (specified by M(s))

is determined accordingly. Such maximum quantization interval
can also be determined at the decoder side, and hence, the said
first bit-plane (specified by M(s)) need not be transmitted as
part of the scalable bitstream in this case (for significant

5   band).

Although the encoding of a digital signal into a scalable
bitstream is described, it shall also be understood that the
invention also includes the decoding of the scalable bitstream

10   into a decoded digital signal by the reverse of the method as
described above.

In one embodiment of the invention, a method for decoding the
scalable bitstream into the digital signal is provided which

15   includes de-multiplexing the scalable bitstream into a core-
layer bitstream and an enhancement-layer bitstream, decoding
and de-quantizing the core-layer bitstream to generate a core-
layer signal, bit-plane decoding the enhancement-layer based on
perceptual information of the digital signal, performing an

20   error mapping based on the bit-plane decoded enhancement-layer
signal and the de-quantized core-layer signal to generate an
reconstructed transformed signal, wherein the reconstructed
transformed signal is the digital signal. It should be noted
that the method for decoding the scalable bitstream may be used

25   in combination with but also separately from the method for
encoding a digital signal into the scalable bitstream as
described above.

The reconstructed transformed signal may be transformed to

30   generate the digital signal, if the digital signal is in a
domain different from the reconstructed transformed signal.

The exact implementation of the decoding of the scalable
bitstream to generate the digital signal depends on how the
scalable bitstream is encoded by the encoder. In one example,
the reconstructed transformed signal may be transformed using
5   intMDCT to generate the digital signal. The core-layer
bitstream may be decoded and de-quantized according to the MPEG
AAC specification. The error mapping is performed by adding the
lower quantization threshold used for de-quantizing the
transformed signal and the bit-plane decoded enhancement-layer
10   bitstream to generate the reconstructed transformed signal. The
advantages and other implementations of the decoder are similar
to the encoder, which has already been described above.

The perceptual information of the digital signal may be
15   obtained by de-multiplexing the scalable bit-stream, if the
perceptual information has been multiplexed into the scalable
bitstream as side information. Alternatively, if the core-layer
bitstream is perceptually encoded, the perceptual information
obtained by decoding and de-quantizing the core-layer bitstream
20   may be used for bit-plane decoding of the enhancement-layer
bitstream.

In an embodiment of the invention, the enhancement-layer
bitstream is bit-plane decoded in a consecutive sequence to
25   generate a plurality of bit-planes comprising a plurality of
bit-plane symbols, and the bit-planes are shifted based on the
perceptual information of the digital signal to generate the
bit-plane decoded enhancement-layer bitstream.

30   In another embodiment of the invention, the enhancement-layer
bitstream is bit-plane decoded in a sequence based on the
perceptual information of the digital signal to generate a

plurality of bit-planes comprising a plurality of bit-plane symbols, thereby generating the bit-plane decoded enhancement-layer bitstream.

5 The perceptual information of the digital signal may be at least one of the following:
-the bit-plane which corresponds to the enhancement-layer bitstream when the bit-plane decoding of the enhancement-layer bitstream starts M(s); and

10 -the Just Noticeable Distortion (JND) level of the digital signal, , wherein s corresponds to a frequency band of the digital signal.

The bit-plane which corresponds to the enhancement-layer

15 bitstream when the bit-plane decoding of the enhancement-layer bitstream starts M(s) is determined from the maximum quantization interval used for de-quantizing the core-layer bitstream.

20 The second aspect of the invention not only relates to a method for decoding a scalable bitstream into a digital signal, but also includes a computer program, a computer readable medium and a device for implementing the said method.

25 Detailed Description of the Invention

Various embodiments and implementations of the invention shall now be described in detail with reference to the figures, wherein:

30

Figure 1 shows an encoder according to an embodiment of the invention.

Figure 2 shows a decoder according to an embodiment of the invention.

5   Figure 3 illustrates a structure of a bit-plane coding process.

Figure 4 shows an encoder according to an embodiment of the invention.

10   Figure 5 shows a decoder according to an embodiment of the invention.

Figure 6 shows an encoder according to an embodiment of the invention.

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Figure 7 shows a decoder according to an embodiment of the invention.

Figure 1 shows an encoder 100 according to an embodiment of the
20   invention.

The encoder 100 serves for generating a scalable bitstream, and comprises two distinct layers, namely, a core-layer which generates the core-layer bit-stream, and a Lossless Enhancement
25   (LLE) layer which generates the enhancement-layer bitstream.

The encoder comprises a domain transformer 101, a quantizer 102, an error mapping unit 103, a perceptual bit-plane coder 104 and a multiplexer 105.

30

In the encoder 100, the digital signal is first transformed by the domain transformer 101 to a suitable domain, such as

frequency domain, resulting in a transformed signal. The
coefficients of the transformed signal are quantized by the
quantizer 102 and coded to generate the core-layer bitstream.
Error-mapping is performed by the error mapping unit 103, which
5 corresponds to the LLE layer, to remove the information from
the coefficients of the transformed signal that has been used
or coded in the core layer to form the core-layer bitstream.
The resultant residual or error signal, specifically error
coefficients, are bit-plane coded by the bit plane coder 104 to
10 generate the embedded LLE bitstream. This embedded bit-stream
can be further truncated to lower bit-rates at the encoder 100
or at a corresponding decoder (as the decoder 200 shown in
figure 2 and described below), or in the communication channel
to meet the rate/fidelity requirements. A perceptual model 106
15 is used to control the bit-plane coding of the error
coefficients, so that the bits of the error coefficients which
are more perceptually significant are coded first.

Finally, the resultant LLE layer bitstream is multiplexed with
20 the core layer bitstream by the multiplexer 105 to generate the
scalable bitstream. In addition, perceptual information for
controlling the bit-plane coding of the error coefficients may
also transmitted as a side information so that a corresponding
bit-plane decoder is able to reconstruct the error coefficients
25 in a correct order.

When the LLE bit-stream is truncated to lower rates, the
decoded signal would be a lossy version of the original input
signal.

30
Figure 2 shows a decoder 200 according to an embodiment of the
invention.

The decoder 200 decodes a scalable bitstream generated by the
encoder 100 to reconstruct the digital signal which was encoded
by the encoder 100.

5

The decoder 200 comprises a domain transformer 201, a de-
quantizer 202, an error mapping unit 203, a perceptual bit-
plane decoder 204 and a de-multiplexer 205.

10 The de-multiplexer 205 receives the scalable bitstream as input
and splits the scalable bitstream into the core-layer bitstream
and the enhancement-layer bitstream as generated by the encoder
100. The core-layer bitstream is decoded and de-quantized by
the de-quantizer 202 to form the core-layer signal. The
15 enhancement-layer bitstream is perceptually bit-plane decoded
based on the perceptual information given by a perceptual model
206 by the perceptual bit-plane decoder 204, and is
subsequently error mapped by the error mapping unit 203 with
the core-layer signal to generate an enhancement-layer signal.
20 The enhancement-layer signal is finally transformed back to the
domain of the digital signal by the domain transformer 201,
resulting in an enhancement-layer transformed signal which is
the reconstructed digital signal.

25

The processing carried out by the encoder 100 and the decoder
200 is explained in detail in the following.

The input signal is normally transformed to the frequency
30 domain by the domain transformer 101 before it is quantized by
the quantizer 102 (which is part of the core-layer encoder) to
generate the core-layer bitstream. Various transform functions

may be used for transforming the input signal to the frequency
domain, such as Discrete Cosine Transform (DCT), Modified
Discrete Cosine Transform (MDCT), integer MDCT (IntMDCT) or
Fast Fourier Transform (FFT).

5

When MPEG-4 AAC encoder is used as the core-layer encoder (for
audio signal), MDCT is commonly used to transform the input
audio signal to the frequency domain, as described in [1]. In
[13], integer MDCT (IntMDCT) is proposed as a revertible
10 approximation to the Modified Discrete Cosine Transform (MDCT)
filterbank used with the MPEG-4 AAC encoder. A generally used
way to implement the IntMDCT is to factorize the MDCT
filterbank into a cascade of Givens rotations in the form of:

$$\begin{pmatrix} \cos\alpha & -\sin\alpha \\ \sin\alpha & \cos\alpha \end{pmatrix},$$

15 which is further factorized into three lifting steps

$$\begin{pmatrix} \cos\alpha & -\sin\alpha \\ \sin\alpha & \cos\alpha \end{pmatrix} = \begin{pmatrix} 1 & \dfrac{\cos\alpha-1}{\sin\alpha} \\ 0 & 1 \end{pmatrix} \begin{pmatrix} 1 & 0 \\ \sin\alpha & 1 \end{pmatrix} \begin{pmatrix} 1 & \dfrac{\cos\alpha-1}{\sin\alpha} \\ 0 & 1 \end{pmatrix}.$$

Each lifting step can be approximated by a revertible integer
to integer mapping with the rounding to the nearest integer
operation $r : \mathbf{R} \rightarrow \mathbf{Z}$. For example, the last lifting step is
20 approximated by:

$$\left(x_1, x_2\right) \rightarrow \left(x_1 + r\left(\frac{\cos\alpha-1}{\sin\alpha} x_2\right), x_2\right),$$

which can be losslessly reverted by:

$$\left(x_1', x_2'\right) \rightarrow \left(x_1' - r\left(\frac{\cos\alpha-1}{\sin\alpha} x_2'\right), x_2'\right).$$

IntMDCT is thus obtained by implementing all the Givens
25 rotations with the revertible integer mapping as described
above.

In the decoder, intMDCT can again be used by the domain
transformer 201 to transform the enhancement-layer signal to
the (reconstructed) digital signal.

5   At the core layer, the coefficients $c(k)$ of the transformed
signal, where $k = 1,...,1024$ is the length of a frame of the
core-layer bitstream, are quantized by the quantizer 102 and
coded into the core-layer bitstream. In the context of an input
audio signal, the transformed signal coefficients may be
10 quantized according to the quantization values of a MPEG-4 ACC
coder, a MPEG-1 Layer 3 Audio (MP3) coder or any proprietary
audio coder.

When the MPEG-4 ACC coder is used in conjunction with the
15 IntMDCT, the transformed signal coefficients (also known as the
IntMDCT coefficients), $c(k)$, are first normalized as

$c'(k) = \alpha \cdot c(k)$

to approximate the normalized outputs to the outputs of the
MDCT filterbank. The normalized IntMDCT coefficients, $c'(k)$,
20 are then quantized and coded, for example, according to an AAC
quantizer (see [19]) which is given as follows:

$$i(k) = \text{sgn}\left[c'(k)\right]\left\lfloor \left(\frac{|c'(k)|}{\sqrt[4]{2^{\text{scale\_factor}(s)}}}\right)^{3/4} + 0.4054 \right\rfloor.$$

Here $\lfloor \cdot \rfloor$ denotes the flooring operation which truncates a
floating operand to integer, $i(k)$ is the AAC quantized
25 coefficients and scale_factor(s) is a scale factor of a scale-
factor band s in which the coefficient $c(k)$ belongs to. The
scale factors can be adjusted adaptively by a noise shaping
procedure so that the quantization noise is best masked by the

masking threshold of the human auditory system. A widely
adopted approach for this noise shaping procedure is the nested
quantization and coding loop as described in details in [1].

5   The quantized coefficients i(k) are noiselessly coded (in this
     example by the quantizer 102), for example, using Huffman code
     or Bit-Sliced Arithmetic Code (BSAC) as described in [17]. BSAC
     is preferred if bit-rate scalability is further required in the
     core layer bitstream. The scale factors are differentially
10   encoded, for example, by the DPCM encoding process described in
     [1], or using Huffman code. The core-layer bitstream can then
     be generated by multiplexing all the coded information
     according to the AAC bitstream syntax.

15   A more comprehensive description on MPEG AAC can be found in
     [1] or in the International standard document on MPEG AAC [19].

     It should be noted that although the mechanism of embedding the
     MPEG-4 AAC compliant bit-stream is described, it is also
20   possible to use bitstreams which are compliant to other coders
     such as MPEG 1/2 Layer I, II, III (MP3), Dolby AC3, or SONY's
     ATRAC proprietary encoders as described in [20].

     When quantizer 102 works according to the MPEG AAC coder, the
25   de-quantizer 202 preferably works according to a MPEG AAC
     decoder for decoding and de-quantizing the core-layer bit-
     stream in the decoder 200. Specifically, the de-quantizer 202
     is used to generate the core-layer signal which is subsequently
     used for error mapping by the error mapping unit 203 in the
30   decoder 200 to generate the enhancement-layer signal as will be
     described below.

However, it should be noted that de-quantizers according to other specifications such as MP3 or other proprietary decoders may be used in the decoder 200.

5

In the LLE layer, an error mapping procedure is employed to remove the information that has already been coded in the core-layer bit-stream. A possible approach to build such an error mapping procedure is by subtracting the lower (closer to zero)
10   quantization threshold of each quantized coefficient from the corresponding transformed input signal coefficient.

This can be illustrated as:

$$e(k) = c(k) - thr(k),$$

15   where the $thr(k)$ is the lower (closer to zero) quantization threshold for $c(k)$, and $e(k)$ is the error coefficient which represents the error signal.

When the MPEG-4 AAC coder is used as the quantizer:

20    $$thr(k) = sgn[c(k)]\left|\frac{\sqrt[4]{2^{scale\_factor(s)}}[i(k) - 0.4054]^{4/3}}{\alpha}\right|.$$

In practical applications, to ensure robust reconstruction, a mapping from integer $i(k)$ to integer $thr(k)$ may be performed using a lookup table. As can be clearly seen from the above formula, a total of 4 tables are required for different values
25   of scale_factors (as the same table can be shared among different values scale_factors if they have a modulus 4 by bit-shifting), wherein each table contains the mapping between all possible values of $i(k)$ and corresponding $thr(k)$ for any scale_factor from the set of those with modulus 4.

It is also possible to perform the error mapping procedure by subtracting a reconstructed coefficient of the transformed input signal coefficient from the transformed signal

5 coefficient as described in [3], which can be illustrated as:

$$e(k) = c(k) - \hat{c}(k)$$

wherein $\hat{c}(k)$ is the reconstructed transformed signal

10 coefficient.

In general, it is also possible to perform the error mapping procedure based using:

15 $$e(k) = c(k) - f(k)$$

wherein $f(k)$ is any function which corresponds to $c(k)$, such as

$$f(k) = \frac{1}{2}\left(thr(k + 1) - thr(k)\right).$$

20

Clearly, for $c(k)$ that has already been significant in the core layer ($thr(k) \neq 0$), the sign of the IntMDCT residual $e(k)$ can be determined from the core layer reconstruction and hence only its amplitude is needed to be coded in the LLE layer. In

25 addition, it is well known that for most audio signals, $c(k)$ can be approximated by Laplacian random variables with the probability density function (pdf):

$$f(c(k)) = e^{-|c(k)|\sqrt{2/\sigma^2}} \Big/ 2\sigma^2 ,$$

where σ is the variance of c(k). From the "memoryless" property of a Laplacian pdf it is easy to verify that the amplitude of e(k) is geometrically distributed as,

$$f\left(\left|e\left(k\right)\right|\right) = \beta \cdot \theta\left(k\right)^{\left|e(k)\right|}, \qquad (1)$$

5   where the distribution parameter θ(k) is determined by the variance of c(k) and the step size of the core layer quantizer. This property enables a very efficient bit-plane coding scheme, such as the bit-plane Golomb code (BPGC) 0 for encoding the error signal to be applied.

10

In the decoder 200, the coefficients of the transformed signal may be reconstructed by the error mapping procedure performed by the error mapping unit 203 according to the following equation:

15

c(k) = e'(k) + thr(k)

wherein e'(k) are the decoded error coefficients which describe the bit-plane decoded enhancement-layer bitstream, which
20   corresponds to the error coefficients e(k) in the encoder 100. Hence it can be seen that with the transformed signal coefficients c(k) can be regenerated from the decoded error coefficients e'(k) (possible a lossy version if the LLE bit-stream is truncated to lower rates) and the quantization
25   threshold thr(k) generated in the same manner in the encoder with the quantization index i(k) contained in the embedded core-layer (AAC) bitstream.

Similar to the encoder 100, the transformed signal coefficients
30   c(k) in the decoder 200 may also be generated using (adding)

the decoded error coefficients e'(k) and reconstructed
coefficients of the core-layer bitstream. Also, the transformed
signal coefficients c(k) may be generated using (adding) the
decoded error coefficients e'(k) and a function of c(k).

To produce the scalable to lossless portion of the final
embedded lossless bit-stream, the residual or error signal is
further coded by the perceptual bit-plane coder 104 using bit-
plane coding, an embedded coding technology that has widely
adopted in audio coding [3] or image coding [5], in the LLE
layer.

A description of a general bit-plane coding procedure can be
found in [4] and [15]. Consider an input n-dimensional data
vector $\underline{x}_n = \{x_1, \ldots, x_n\}$ where x_i is extracted from some random
sources of some alphabet $A \subset \mathfrak{R}$. Clearly, x_i can be
represented in a binary format

$$x_i = (2s_i - 1) \cdot \sum_{j=\infty}^{-\infty} b_{i,j} \cdot 2^j, i = 1, \ldots, k$$

by cascading of binary bit-plane symbols that comprises of a
sign symbol

$$s_i \overset{\Delta}{=} \begin{cases} 1 & x_i \geq 0 \\ 0 & x_i < 0 \end{cases},$$

and amplitude symbols $b_{i,j} \in \{0, 1\}$. In practice, the bit-plane
coding could be started from the maximum bit-plane M of vector
$\underline{x}_n$ where M is an integer that satisfies

$$2^{M-1} \leq \max\left\{|x_i|\right\} < 2^M, i = 1, \ldots, k,$$

and stopped at bit-plane 0 if $\underline{x}_n$ is a integer vector.

The bit-plane coding and decoding process according to one embodiment of the invention and as for example performed by the perceptual bit-plane coder 104 and the perceptual bit-plane decoder 204 is explained in the following with reference to figure 3.

Figure 3 illustrates a structure of the above bit-plane coding (BPC) process, where each input vector is first decomposed into the binary sign and amplitude symbols, which are then scanned, in a desired order, by a bit-plane scanning unit 301 and coded by an entropy coder 302 (e.g. as arithmetic code, as Huffman code or as run-length code). In addition, a statistical model 303, for example based on a Laplacian distribution of the input signal, is usually used to determine the probability assignment for each binary symbol to be coded. In the corresponding decoder, the data flaw is reversed, i.e. the output of the entropy encoder 302 is decoded by an entropy decoder 303 using a corresponding statistical model 304 and the result is used by a bit-plane reconstruction unit 304 to rebuild the bit-plane, where the sign and amplitude symbols which are decoded to rebuild the bit-plane of the data vector follows the same scanning order in the encoder.

The most significant advantage of having a bit-plane coding system as above is the resulted compression bit-stream can be easily truncated to any desirable rates, where a reproduction

data vector $\hat{x}$ can still be obtained by a partial reconstructed bit-planes decoded from this truncated bit-stream. For best coding performance, an embedded principle (see [24]) is usually adopted in BPC, according to which the bit-plane symbols are coded in the order of decreasing rate-distortion slope so that symbols with most signification contribution to the final distortion per unit rate are always coded first.

The selection of the order of bit-plane scanning depends on the desired distortion measurement. When the mean square error (MSE) or the expectation on the square error function is used as the distortion measurements as shown:

$$d\left(\underline{x}_n, \hat{\underline{x}}_n\right) = \frac{1}{n} \sum_{i=1}^{n} \left(x_i - \hat{x}_i\right)^2$$

wherein $d\left(\underline{x}_n, \hat{\underline{x}}_n\right)$ is the distortion value, x_n is the original data vector, and $\hat{\underline{x}}_n$ is the reconstructed vector of x_n at the decoder. Results from [24] shows that the embedded principle is satisfied well by a sequential bit-plane scanning and coding procedure for most sources, except those with very skew bit-plane symbols distribution.

An example of a simple sequential bit-plane scanning and coding procedure comprises the following steps:

1. Starting from the most significant bit-plane $j = M - 1$;
2. Encode only $b_{i,j}$ with $b_{i,M-1} = b_{i,M-2} = \cdots = b_{i,j+1} = 0$. If $b_{i,j} = 1$ in the significance scan, encode s_i; (significance pass);

3. Encode $b_{i,j}$ that are not be encoded in the significance pass

(Refinement pass);

4. Progress to bit-plane $j - 1$.

5   List 1. Bit-plane scanning & coding procedure

The above procedure is iterated until certain terminating
criterion, which is usually a pre-defined rate/distortion
constraint, is reached. In addition, further adjustment of
10 coding sequence in a significance pass may be required if bit-
plane symbols are found to have unequal distributions.

An example of the above sequential coding procedure is
illustrated by considering a data vector x with dimension 4,
15 say {9,-7,14,2}. So it is bit-plane coded from its most
significant bit-plane 4. The significance pass is begun with
since all elements are insignificant yet. (X denotes the bypass
symbols). The sign is coded as follows: positive is coded as 1,
and negative is coded as 0.

20

Data vector	9	-7	14	1
1st significant pass (sign)	1(sign: 1)	0	1(sign:1)	0
1st refinement pass	X	X	X	X
2nd significant pass (sign)	X	1(sign:0)	X	0
2nd refinement pass	0	X	1	X
3rd significant pass	X	X	X	0

3rd refinement pass	0	1	1	X
4th significant pass	X	X	X	1(sign:1)
4th refinement pass	1	1	0	X

Thus the output binary stream is 11011010001001111110, which is then entropy coded and sent to the decoder. At the decoder, the bit-plane structure of the original data vector is

5   reconstructed. If the entire binary stream is received by the decoder, the bit-plane of the original data vector can be restored and thus, a lossless reconstruction of the original data vector is obtained. If only a subset (most significant part) of the binary stream is received, the decoder still able

10   to restore a partial bit-plane of the original data vector, so that a coarse reconstruction (quantized) version of the original data vector.

The above is only a simple example of bit-plane scanning and

15   coding procedure. In practice, the significant pass can be further fractionized to explore the statistical correlation of elements in the data vector, such as the bit-plane coding process in JPEG2000, or that in the embedded audio coder (EAC) described in [4].

20

The above sequential bit-plane scanning and coding procedure only provides an effort to optimize the MSE performance. In area of audio, image or video coding, minimizing perceptual distortion instead of MSE is normally a more efficient coding

25   method for obtaining optimal perceptual quality in reconstructed audio, image or video signal. Therefore, the

sequential bit-plane coding of the error signal is definitely a
sub-optimal option.

In the encoder 100, the error coefficients are preferably
5   grouped into frequency bands so that each frequency band s
contains a number of error coefficients in consecutive order.
(The scale factor band grouping may be based on the band
grouping adopted in the quantizer 102 if a perceptual coder is
used as the quantizer 102. However other band grouping is also
10   possible).

A frequency band s is said to be significant if there exist an
error coefficient in the frequency band s such that the
quantized coefficient thr(k) from the quantizer is not zero. In
15   other words, if $e(k)$ is an error coefficient in frequency band
s:

$$e(k) = c(k) - thr(k),$$

20   frequency band s is significant is $thr(k) \neq 0$ ($thr(k) = 0$ when
$i(k) = 0$), and hence $e(k) = c(k)$, else frequency band s is
considered insignificant.

Perceptual significance of bits of the error coefficients can
25   be determined by the level of Just Noticeable Distortion (JND)
at a frequency location i. This level of JND, T_i, can be
determined from a perceptual model such as psychoacoustic model
(I or II) or any proprietary perceptual models. When a
perceptual quantizer is used for forming the core-layer
30   bitstream, the perceptual model used in the quantizer may also

be used to generate the JND for perceptual bit-plane coding of the error coefficients.

For simplicity, perceptual significance of bits of the error
5 coefficients in a same frequency band s can be set to the same value.

In the following, a possible implementation of perceptual bit-plane coding is explained with reference to Fig.4.
10
Figure 4 shows an encoder 400 according to an embodiment of the invention.

Analogously to the encoder 100, the encoder 400 comprises a
15 domain transformer 401, a quantizer 402, an error mapping unit 403, a perceptual bit-plane coder 404 (using a perceptual model 406) and a multiplexer 405.

The perceptual BPC block, i.e., the perceptual bit-plane coder
20 404 comprises a bit-plane shifting block 407 and a conventional BPC block 408.

In the bit-plane shifting block 407, the bit-planes are perceptually shifted, and the perceptually shifted bit-planes
25 are coded in the BPC block 408 in a conventional sequential scanning and coding manner.

Consider the following (modified) perceptually weighted distortion measurement
30

$$d\left(\underline{x}_n, \underline{\hat{x}}_n\right) = \frac{1}{n} \sum_{i=1}^{n} \left(x_i - \hat{x}_i\right)^2 w_i\left(x_i\right).$$

In the context of perceptual audio coding, the audio signal is usually quantized and coded in the frequency domain so that the data vector $\underline{x}_n$ is the transformed audio signal and the weighting function $w_i\left(x_i\right)$ is the importance of x_i at different frequency locations i, i.e.,

$$w_i\left(x_i\right) = \frac{1}{T_i}.$$

The above perceptually weighted distortion function may be re-written as follows:

$$d\left(\underline{x}_n, \underline{\hat{x}}_n\right) = \frac{1}{n} \sum_{i=1}^{n} \frac{1}{T_i} \left(x_i - \hat{x}_i\right)^2 = \frac{1}{n} \sum_{i=1}^{n} \left(\frac{1}{\sqrt{T_i}} x_i - \frac{1}{\sqrt{T_i}} \hat{x}_i\right)^2$$

$$= \frac{1}{n} \sum_{i=1}^{n} \left(x_i' - \hat{x}_i'\right)^2,$$

where

$$x_i' \overset{\Delta}{=} \frac{1}{\sqrt{T_i}} x_i, i = 1, \ldots, n.$$

Hence the weighting square error function now becomes a square error function on the scaled vector $\underline{x}_n' = \left\{x_1', \ldots, x_n'\right\}$. Therefore, perceptually optimized coding of $\underline{x}_n$ can be achieved by simply performing sequential bit-plane coding on $\underline{x}_n'$. In the corresponding decoder, each element of the bit-plane decoded

data vector $\underline{\hat{x}}'$ can be scaled back to obtain a reconstructed
data vector $\underline{\hat{x}}$ as follows.

$$\hat{x}_i = \sqrt{\hat{T}_i} \cdot \hat{x}'_i, \, i = 1, \ldots, n.$$

Clearly, the weights T_i are preferably transmitted to the
decoder as side information if they are unknown in the decoder.

w_j is further quantized to an even integer power of 2 so that
it becomes

$$\hat{T}_i = 2^{2\tau_i},$$

where

$$\tau_i = \frac{1}{2} \log_2 T_i$$

and the scaled data vector can thus be obtained by bit-shifting
each element in the original data vector as follows

$$x'_i = 2^{-\tau_i} x_i.$$

which is easily obtained by performing right shifting operation
on x_i by τ_i. For example, if x_i = 00010011 and τ_i = -2, the
scaled data vector element x'_i is then 01001100; if τ_i = 2 it
will become 00000100.11.

In this way, the bit-planes of the error coefficients are
perceptually shifted in a manner such that when a sequential

bit-plane coding is performed on the shifted bit-plane, bits
which are more perceptually significant (instead of having the
highest MSE) can be encoded first.

5   Clearly, if each element in the original data vector is integer
with limited word length, e.g., if each element in $\underline{x}$ has a
maximum bit-plane of L, lossless coding of $\underline{x}$ can be achieved
if every x_i' in the scaled vector is bit-plane coded from bit-
plane $-\tau_i \sim L - \tau_i$.

10

As mentioned earlier, information on the perceptual
significance such as the level of JND can be provided to the
bit-plane shifting block from a perceptual model.

15   In the bit-plane coding process, a maximum bit-plane, M(s), can
be used to specify the starting bit-plane at which the bit-
plane scanning and coding should start. The maximum bit-plane
M(s) and τ_i should preferably be transmitted as side
information in the scalable bitstream to the corresponding
20   decoder in order for the decoder to be able to decode the
bitstream correctly. To reduce the amount of side information,
M(s) and τ_i may be constrained to the same value for the same
scale factor band s in the encoder.

25   The value of the maximum bit-plane M(s) in each frequency band
s can be determined from the error coefficients e(k) using the
following expression:

$$2^{M(s)-1} \leq \max \left(\left| e\left(k\right) \right| \right) < 2^{M(s)}, k \in s.$$

30

Furthermore, the maximum absolute value of the error

coefficients $\max\left(\left|e\left(k\right)\right|\right)$ in each significant frequency band s

are bounded by the quantizer interval of the perceptual

quantizer:

5

$$\max\left(\left|e\left(k\right)\right|\right) \leq \text{thr}\left(i\left(k\right) + 1\right) - \text{thr}\left(i\left(k\right)\right).$$

Therefore, this results in the maximum bit-plane M(s) for each

significant frequency band s to be determined from the

10   following expression:

$$2^{M(s)-1} \leq \max\left(\left|\text{thr}\left(i\left(k\right) + 1\right) - \left|\text{thr}\left(i\left(k\right)\right)\right|\right|\right) < 2^{M(s)}, k \in s.$$

Since the quantized coefficients of the perceptual quantizer

15   i(k) is known to the decoder, it is thus not necessary to

transmit the maximum bit-plane M(s) as side information to the

decoder for the significant frequency bands s.

The value of the maximum bit-plane M(s) may also be predefined

20   in the encoder and decoder, and hence, need not be transmitted

as the side information.

Figure 5 shows a decoder 500 according to an embodiment of the

invention.

25

The decoder 500 implements a perceptual bit-plane decoder

which comprises bit-plane shifting and conventional (sequential)

bit-plane coding.

Analogously to the decoder 200 the decoder 500 comprises a
domain transformer 501, a de-quantizer 502, an error mapping
unit 503, a perceptual bit-plane decoder 504 (using a
perceptual model 506) and a de-multiplexer 505.

Similar to the perceptual bit-plane coder 404, the perceptual
bit-plane decoder 504 comprises a bit-plane shifting block 507
and a conventional BPC block 508.

The enhancement-layer bitstream generated by the encoder 400 is
bit-plane decoded by the decoder 500 in the consecutive
sequential manner (same sequential bit-plane scanning procedure
as the encoder 400) to reconstruct the bit-planes. The
reconstructed bit-planes are shifted in the reverse manner of
the encoder 400, based on the received or regenerated value τ_i,
to generate the decoded error coefficients e'(k) which describe
the bit-plane decoded enhancement-layer bitstream.

Figure 6 shows an encoder 600 according to an embodiment of the
invention.

The encoder 600 uses perceptual bit-plane coding.

The encoder 600 comprises a domain transformer (intMDCT) 601, a
quantizer (ACC quantizer and coder) 602, an error mapping unit
603, a perceptual significance calculation unit 604 (using a
psychoacoustic model 605), a perceptually bit-plane coding unit
606 and a multiplexer 607.

In this implementation, the scanning order of the bit-planes
and the bit-plane symbols need not be sequential, but based on
perceptual importance of the bit-plane symbols corresponding to

different frequency bands. The perceptual importance of the bit-plane symbols is determined by calculating parameters related to the perceptual information, such as perceptual significance and first (maximum) bit-plane for bit-plane
5   decoding. The calculation of the perceptual information parameters is represented as the perceptual significance calculation block, i.e., the perceptually bit-plane coding unit 604.

10 There are numerous ways to determine the perceptual importance, or specifically the perceptual significance, of the bit-plane symbols corresponding to different frequency bands. One widely adopted way is by using the psychoacoustics model, such as the Psychoacoustics Model 2 described in [19], of the input digital
15 signal. The just noticeable distortion (JND) level T(s) for each frequency band determined using the psychoacoustic model can be converted to the unit of bit-plane level τ(s) as follows:

$$\tau(s) = \frac{1}{2} \log_2 \left(T(s) \right)$$

20

However, this invention does not constrain the method on how T(s) or τ(s) can be obtained.

Now let Ps(s) represent the perceptual significance of
25 frequency band s, which can be determined by the distance from M(s) to τ(s) as,

Ps(s) = M(s) − τ(s).

It can be further noted that the noise level, or the level of
the IntMDCT error coefficients e(k) would tend to be flat with
respect to the JND level for significant bands (as a result of
the noise shaping mechanism in the core coder). In other words,
5   the value of Ps(s) would be very close, if not identical, for
significant frequency bands. This fact can be explored in the
method according to the invention by sharing a common factor
Ps_common for all the significant bands. Possible selections of
Ps_common can be the average value, the maximum value, the
10 minimum value, or any other reasonable function of Ps(s) for
all s that are significant. The Ps(s) can then be normalized as
follows:

$$Ps'(s) = Ps(s) - Ps\_common,$$

15

Since it is known that for significant band s, Ps'(s) would be
zero, and therefore, need not be transmitted to the decoder.
Otherwise, for insignificant band s, Ps'(s) should preferably be
transmitted to the corresponding decoder as side information.
20

In some other examples when there is no significant band,
Ps_common can be set to 0.

It is also possible to use the noise shaping procedure in the
25 core encoder to cater to the need for perceptual coding. Hence
there is no need to further implement any noise shaping, or
perceptual significant identification in the enhance layer. In
such cases, Ps'(s) = 0 can be set for all s. Usually they do not
need to be transmitted to the decoder if it is known by the
30 decoder that they are all zero.

A possible implementation of the perceptual bit-plane coding
mechanism can be described using the following pseudo code.
Here the total number of the frequency band is denoted as
s_total.

5

1. Find frequency band s with largest $Ps'(s)$

2. Encode bit-plane symbols of bit-plane M(s) for e(k) in band s

3. M(s) = M(s) -1; $Ps'(s) = Ps'(s) - 1$

4. if there exists band s for which $M(s) \geq 0$ goto 1.

10

A method for obtaining the maximum bit-plane M(s) is described
here.

For significance band, M(s) can be determined from the maximum

15 quantization interval of the quantizer if a perceptual
quantizer such as an AAC quantizer is used. Specifically, M(s)
is an integer that satisfies:

$$2^{M(s)-1} \leq \max\left(\left\|\text{thr}\left(i\left(k\right)+1\right) - \left|\text{thr}\left(i\left(k\right)\right)\right|\right\|\right) < 2^{M(s)}, k \in s.$$

20

In this case, M(s) does not need to be transmitted to the
decoder as i(k) would be known to the decoder.

For insignificant bands M(s) can be calculated from e(k) as

25 follows:

$$2^{M(s)-1} \leq \max\left(\left\|e\left(k\right)\right\|\right) < 2^{M(s)}, k \in s,$$

and for those bands, M(s) should preferably be sent to the
decoder as side information such an information is not
contained in the core-layer bit-stream.

5   The value of the maximum bit-plane M(s) may also be predefined
in the encoder 600 and the corresponding decoder, and hence,
need not be transmitted as the side information.

Other alternative approaches to explore parameter Ps(s) in a
10   bit-plane coding approach, towards some desired noise shaping
goals, are also possible. In general, Ps(s) can also be
obtained by any functions of M(s) and τ(s), for example the
following:

15   $Ps(s) = M(s) - 2\tau(s)$, or

$$Ps(s) = \frac{M(s) - \tau(s)}{2}.$$

Figure 7 shows a decoder 700 according to an embodiment of the
invention.
20

The decoder 700 is the corresponding decoder of the coder 600,
wherein the perceptual bit-plane decoding is implemented using
the perceptual bit-plane scanning procedure as described above.

25   The decoder 700 accordingly comprises a domain transformer
(reverse intMDCT) 701, a de-quantizer (ACC de-quantizer and
decoder) 702, an error mapping unit 703, a perceptual
significance calculation unit 704, a perceptually bit-plane
decoding unit 706 and a de-multiplexer 707.
30

In the decoder 700, for significant band, $Ps'(s)$ is set to zero, and $M(s)$ can be calculated from the AAC quantization index $i(k)$ in the same manner in the encoder, i.e.:

$$5 \quad 2^{M(s)-1} \leq \max\left(\left|\left|thr\left(i(k)+1\right)-\left|thr\left(i(k)\right)\right|\right|\right|\right) < 2^{M(s)}, k \in s.$$

For insignificant band, $Ps(s)$ and $M(s)$ can simply be recovered from the transmitted side information. Once $Ps(s)$ and $M(s)$ is recovered for all frequency bands, the IntMDCT error coefficients $\hat{e}(k)$ can be easily reconstructed by decoding the received bit-stream and reconstruct its bit-plane symbols in a order that is exact the same as that in the encoder 700. For example, the decoding process for the encoding example given above would be:

1. Find frequency band s with largest $Ps'(s)$
2. Decode bit-plane symbols of bit-plane $M(s)$ for $\hat{e}(k)$ in band
3. $M(s) = M(s)-1$; $Ps'(s) = Ps'(s) - 1$
4. If there exists band s for which $M(s) \geq 0$ goto 1.

Determining the maximum bit-plane for bit plane coding of error coefficients.

For a significant band s (i.e., the error coefficient $e(k) \neq c(k)$ or $\exists k \in s, i(k) \neq 0$), the maximum absolute value of $e(k)$ is bounded by the quantizer interval in the AAC quantizer as:

$$\max\left(\left|e(k)\right|\right) \leq thr\left(i(k)+1\right) - thr\left(i(k)\right).$$

Therefore, the maximum bit-plane M(k) can be determined using:

$$2^{M(k)-1} \leq \max\left(\left|\left|thr\left(i\left(k\right)+1\right)-\left|thr\left(i\left(k\right)\right)\right|\right|\right|\right) < 2^{M(k)}, k \in s.$$

As i(k) is already known by the decoder, M(k) need not be transmitted to the decoder since the decoder is able to regenerate thr(k) and hence, M(k) from i(k) for significant band s.

For insignificant band, M(k) can be calculated from e(k) as follows:

$$2^{M(s)-1} \leq \max\left(\left|e\left(k\right)\right|\right) < 2^{M(s)}, k \in s,$$

and the calculated M(s) is preferably transmitted with the enhancement-layer bitstream as side information for the enhancement-layer bitstream to be bit-plane decoded correctly.

To reduce the amount of side information, M(k) can be further constrained to have the same values for k for the same scale factor band s in the core-layer quantizer. Therefore, M(k) may be denoted also as M(s).

In the decoder 700, the error coefficients corresponding to the error signal can be reconstructed by bit-plane decoding of the enhancement-layer bitstream using the same bit-plane scanning procedure as the encoder based on M(s). For significant band, M(s) can be regenerated using the following:

$$2^{M(k)-1} \leq \max\left(\left\|\mathrm{thr}\left(i\,(k)\,+\,1\right)\,-\,\left|\mathrm{thr}\left(i\,(k)\right)\right\|\right) < 2^{M(k)}, k \in s.$$

For insignificant band, the decoder makes use of the M(s) which is transmitted by the encoder as side information.

5

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